SARDAR RAJA COLLEGE OF ENGINEERING, ALANGULAM

DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGINEERING

MICRO LESSON PLAN

SUBJECT NAME	**:**	**DIGITAL SIGNAL PROCESSING**
SUBJECT CODE	**:**	**EC2302**
YEAR/SEM	**:**	**III YEAR / V SEM**
BRANCH	**:**	**ECE**

STAFF: Ms. THANYA T, Asst.Prof,

DEPT. OF ECE

EC2302 DIGITAL SIGNAL PROCESSING

STAFF NAME: Ms.T.THANYA

AP / ECE

SUBJECT DESCRIPTION AND OBJECTIVE

SUBJECT DESCRIPTION:

The goal of DSP is usually to measure, filter and/or compress continuous real-world analog signals. The first step is usually to convert the signal from an analog to a digital form, by sampling and then digitizing it using an analog-to-digital converter (ADC), which turns the analog signal into a stream of numbers. However, often, the required output signal is another analog output signal, which requires a digital-to-analog converter (DAC). Even if this process is more complex than analog processing and has a discrete value range, the application of computational power to digital signal processing allows for many advantages over analog processing in many applications, such as error detection and correction in transmission as well as data compression.

DSP includes subfields like: audio and speech signal processing, sonar and radar signal processing, sensor array processing, spectral estimation, statistical signal processing, digital image processing, signal processing for communications, control of systems, biomedical signal processing, seismic data processing, etc. Today there are additional technologies used for digital signal processing including more powerful general purpose microprocessors, field-programmable gate arrays (FPGAs), digital signal controllers (mostly for industrial apps such as motor control), and stream processors, among others.

OBJECTIVE:

- To study DFT and its computation
- To study the design techniques for digital filters
- To study the finite word length effects in signal processing
- To study the non-parametric methods of power spectrum estimations
- To study the fundamentals of digital signal processors.

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UNIT I DISCRETE FOURIER TRANSFORM 9

DFT and its properties, Relation between DTFT and DFT, FFT computations using Decimation in time and Decimation in frequency algorithms, Overlap-add and save methods

UNIT II INFINITE IMPULSE RESPONSE DIGITAL FILTERS 9

Review of design of analogue Butterworth and Chebyshev Filters, Frequency transformation in analogue domain – Design of IIR digital filters using impulse invariance technique – Design of digital filters using bilinear transform – pre warping – Realization using direct, cascade and parallel forms.

UNIT III FINITE IMPULSE RESPONSE DIGITAL FILTERS 9

Symmetric and Antisymmetric FIR filters – Linear phase FIR filters – Design using Hamming, Hanning and Blackmann Windows – Frequency sampling method – Realization of FIR filters – Transversal, Linear phase and Polyphase structures.

UNIT IVFINITE WORD LENGTH EFFECTS 9

Fixed point and floating point number representations – Comparison – Truncation and Rounding errors - Quantization noise – derivation for quantization noise power – coefficient quantization error – Product quantization error - Overflow error – Roundoff noise power - limit cycle oscillations due to product roundoff and overflow errors – signal scaling

UNIT V MULTIRATE SIGNAL PROCESSING 9

Introduction to Multirate signal processing-Decimation-Interpolation-Polyphase implementation of FIR filters for interpolator and decimator -Multistage implementation of sampling rate conversion- Design of narrow band filters - Applications of Multirate signal processing.

L: 45, T: 15, Total= 60 Periods

TEXT BOOKS:

1. John G Proakis and Manolakis, " Digital Signal Processing Principles, Algorithms and Applications", Pearson, Fourth Edition, 2007.
2. S.Salivahanan, A. Vallavaraj, C. Gnanapriya, Digital Signal Processing, TMH/McGraw Hill International, 2007

REFERENCES:

1. E.C. Ifeachor and B.W. Jervis, " Digital signal processing – A practical approach", Second edition, Pearson, 2002.
2. S.K. Mitra, Digital Signal Processing, A Computer Based approach, Tata McGraw Hill, 1998.
3. P.P.Vaidyanathan, Multirate Systems & Filter Banks, Prentice Hall, Englewood cliffs, NJ,1993.
4. Johny R. Johnson, Introduction to Digital Signal Processing, PHI, 2006.

MICRO-LESSON PLAN

Week	Hour No.	LECTURE TOPICS	Book
	UNIT I DISCRETE FOURIER TRANSFORM		
I	1	Introduction	T2
	2,3	DFT and its properties	T2
	4,5	Relation between DTFT and DFT	T2
	6	FFT computations using Decimation in time algorithms **(A/V)**	T2
II	7,8	FFT computations using Decimation in frequency algorithms	T2
	9	Overlap-add and save methods	T2
	10	Problems	T2
	11	Problems	T2
	12	Problems	T2
	UNIT II INFINITE IMPULSE RESPONSE DIGITAL FILTERS		
III	13	Introduction	T2
	14,15	Review of design of analogue Butterworth Filters	T2
	16	Review of design of analogue Chebyshev Filters	T2
	17	Frequency transformation in analogue domain	T2
	18	Design of IIR digital filters using impulse invariance technique	T2
IV	19	Design of digital filters using bilinear transform	T2
	20	pre warping	T2
	21	Realization using direct, cascade and parallel forms. **(A/V)**	T2
	22	Problems	T2
	23	Problems	T2
	24	Problems	T2
	UNIT III FINITE IMPULSE RESPONSE DIGITAL FILTERS		
V	25	Introduction	T2
	26	Symmetric and Antisymmetric FIR filters	T2
	27	Linear phase FIR filters	T2
	28,29	Design using Hamming, Hanning and Blackmann Windows	T2

V	30	Frequency sampling method	T2
VI	31	Realization of FIR filters	T2
	32,33	Transversal, Linear phase and Polyphase structures	T2
	34	Problems	T2
	35	Problems	T2
	36	Problems	T2
	UNIT IV FINITE WORD LENGTH EFFECTS		
VII	37	Introduction	T2
	38	Fixed point and floating point number representations	T2
	39	Comparison – Truncation and Rounding errors	T2
	40	Quantization noise – derivation for quantization noise power **(A/V)**	T2
	41	coefficient quantization error – Product quantization error	T2
	42	Overflow error	T2
VIII	43	Round off noise power	T2
	44	limit cycle oscillations due to product round off and overflow errors	T2
	45	signal scaling	T2
	46	Problems	T2
	47	Problems	T2
	48	Problems	T2
	UNIT V MULTIRATE SIGNAL PROCESSING		
IX	49	Introduction	T1
	50	Multirate signal processing	T1
	51	Decimation	T1
	52	Interpolation	T1
	53,54	Polyphase implementation of FIR filters for interpolator and decimator	T1
X	55	Multistage implementation of sampling rate conversion	T1
	56	Design of narrow band filters	T1
	57	Applications of Multirate signal processing (A/V)	T1
	58	Problems	T1
	59	Problems	T1
	60	Problems	T1